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Simulation of NLMS Adaptive Filter for Noise Cancellation

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***Abstract*— Adaptive noise Cancellation (ANC) is an alternative technique for estimation of noise & interference that corrupted the signal. The main objective of the noise cancellation is to estimate the noise signal and to subtract it from original input signal plus noise signal and hence to obtain the noise free signal. There is an alternative method called adaptive noise cancellation for estimating an input signal corrupted by an additive noise. The reference input is adaptively filtered and subtracted from the primary input signal to obtain the estimated signal. In this method the desired signal corrupted by an additive noise can be recovered by an adaptive noise canceller using NLMS (normalized least mean square) algorithm. Estimate the adaptive filter using MATLAB/SIMULINK environment. In the simulation, additive white Gaussian noise is added to the randomly generated information signal and efficiently reduced this noise with minimum or no error by using evolutionary computation with NLMS (normalized least mean square) algorithm.**

***Index Terms*—Adaptive Algorithms, Adaptive Filter, ANC, LMS, NLMS.**

I. INTRODUCTION

Linear filtering is required in a variety of application. A filter will be optimal only if it designed with some knowledge about the input data. If this information is not known, then adaptive filters are used. The adjustable parameters in the filter are assigned with values based on the estimated statistical nature of the signals. So, these filters are adaptable to the changing environment. Adaptive filtering finds its application in noise cancelling, line enhancing, frequency tracking, channel equalization, etc [6].

Variable parameters (Convergence, Tracking, Computational Complexity, Steady state error & Stability) that controls transfer function of an adaptive filter is a system with a linear filter and a means to adjust those parameters according to an optimization algorithm. Because of the complexity of the optimization algorithms, most adaptive filters are digital filters.

An adaptive filter is defined by four aspects:

- The signals being processed by the filter.
- The structure that defines how the output signal of the filter is computed from its input signal.
- The parameters within this structure that can be iteratively changed to alter the filter's input-output relationship.
- The adaptive algorithm that describes how the parameters are adjusted from one time instant to the next.

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Adaptive filter automatically adjusts the parameters of the system to achieve optimal performance according to some criteria. Adaptive filters are having wide range of applications such as noise cancellation, System identification, channel equalization and beam forming etc [8]. The block diagram of adaptive filter is shown in Fig (a).

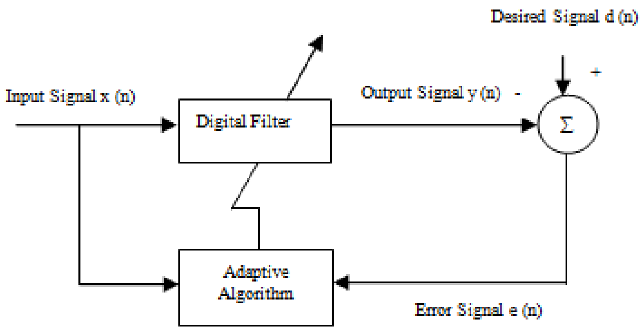

Fig (a) Block diagram of Adaptive Filter

II. ADAPTIVE ALGORITHMS

A. *LMS algorithm*

The properties of Least Mean Square (LMS) algorithm also that makes it the best choice for many real-time systems are simplicity and ease of implementation. The LMS algorithm is introduced by Widrow & Hoff in 1959. Simple, no matrices calculation involved in the adaptation. The Least Mean Square, or LMS, algorithm is a stochastic gradient algorithm that iterates each tap weight in the filter in the direction of the gradient of the squared amplitude of an error signal with respect to that tap weight. The LMS is an approximation of the steepest descent algorithm, which uses an instantaneous estimate of the gradient vector. Adaptive process containing two input signals (1) Filtering process, producing output signal. (2) Desired signal. Adaptive process: Recursive adjustments of filter tap weights.

B. *Normalized LMS algorithm*

The normalized LMS (NLMS) algorithm is a modified form of the standard LMS algorithm. The NLMS algorithm updates the coefficients of an adaptive filter by using the following equation:

$$\bar{w}(n+1) = \bar{w}(n) + \mu e(n)\frac{\bar{u}(n)}{\|\bar{u}(n)\|^2} \qquad (1)$$

This form can be rewritten as,

$$\bar{w}(n+1) = \bar{w}(n) + \mu(n)e(n)\bar{u}(n) \qquad (2)$$

where $$\mu(n) = \mu \|\bar{u}(n)\|^2 \qquad (3)$$

In the previous equation, the NLMS algorithm becomes the same as the standard LMS algorithm except that the NLMS algorithm has a time-varying step size $\mu(n)$. This step size can improve the convergence speed of the adaptive filter. The NLMS algorithm is a potentially faster converging algorithm compared to the LMS algorithm which may come at a price of greater residual error. The main drawback of the pure LMS algorithm is that it is sensitive to the scaling of its Input x (n). This makes it very hard to choose a learning rate µ that guarantees stability of the algorithm. The NLMS is a variant of the LMS algorithm that solves this problem by normalizing with the power of the input.

III. Adaptive Noise Cancellation

The primary aim of an adaptive noise cancellation algorithm is to allow the noisy signal through a filter which suppresses the noise without disturbing the desired signal [1]. The basic block diagram is given in Fig (b).

Adaptive filter system has two inputs, first is the primary input and other is reference signal. The primary input is d (n), which represents the desired signal corrupted with undesired noise and the reference signal x (n), which is the undesired noise to be filtered out of the system. The goal of adaptive filtering system is to reduce the noise portion, and to obtain the uncorrupted desired signal. In order to achieve this, a reference of the noise signal is needed and is called reference signal x (n). However, the reference signal is typically not the same signal as the noise portion of the primary amplitude, phase or time. Therefore the reference signal cannot be simply subtract from the primary signal to obtain the desired portion at the output. In general, noise that affects the speech signals can be modeled using any one of the following: 1. White noise, 2. Colored noise.

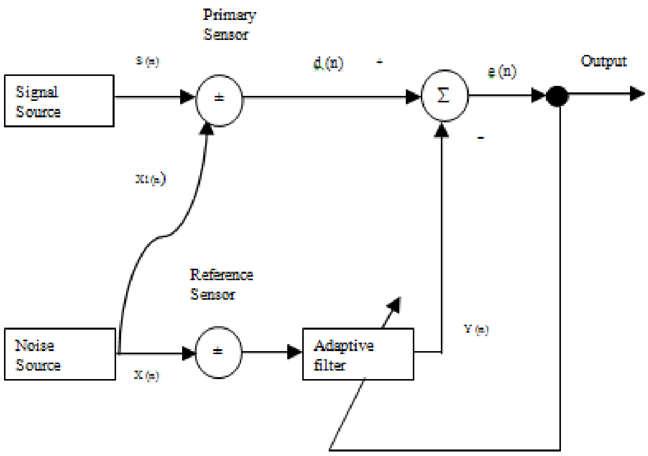

Fig (b) Adaptive Noise Cancellation System

IV. RESULTS AND DISCUSSION

A. Simulink Model

The contents of the journal are peer-reviewed and archival. The journal I In this part evaluation of the performance of LMS adaptive noise cancellation is discussed. Environmental noise polluted sinusoidal signal is extracted. Noise signal is modeled as Gaussian noise. The two signals were added and subsequently fed into the simulation of LMS adaptive filter. The test block diagram of the noise canceller in Simulink is shown in Fig (c).

System inputs are analog signal and Gaussian noise signal. The system outputs are the sinusoidal signal after filtering. Comparisons are worked out in the form of figures, which show the input, desired and error signals. By using manual switch LMS Adaptive filter Step size parameter is changed between high and low constant values. If the step size parameter is at higher constant value the response is fast but showing less accurate and if step size factor is at lower constant value the response may be slow but it is showing more exact performance.

Error Signal is define the difference of desired Signal and Input Signal which is obtained from output port of LMS block. The Signal from this port is the Filtered Signal from which Noise has been adaptively removed out. Single Output Device block is used and signals from output ports of LMS blocks are changed.

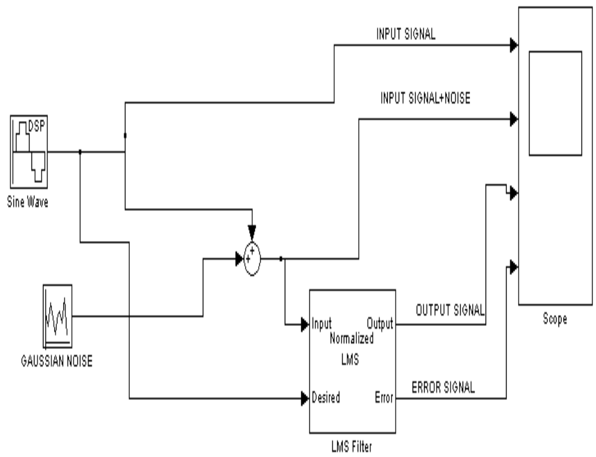

Fig (c) LMS Adaptive Noise Canceller Simulink model

B. Result

Fig (d) is the Adaptive noise canceller Simulink Model scope output. In this the first waveform is represents input signal entering into the system. Second waveform is shows resultant signal in voltage directs after getting corrupted by noise thus generated. Third waveform is represents filter output signal. The fourth waveform is shows the error signal which defined by the difference of desired signal and input signal.

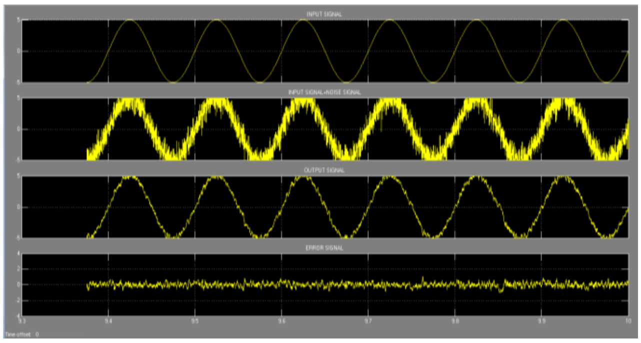

Fig (d) Scope output of adaptive noise canceller Simulink Model Input Signal, Noise Corrupted Signal Output Signal and Error Signal.

C. Discussion

The simulation results of the noise cancellation with LMS Filter as well as the proposed technique are presented. The fig (e) shows the error signal. The variation of error signal after filtration varies between a maximum of -1 to +1 which is well inside the limits to be considered.

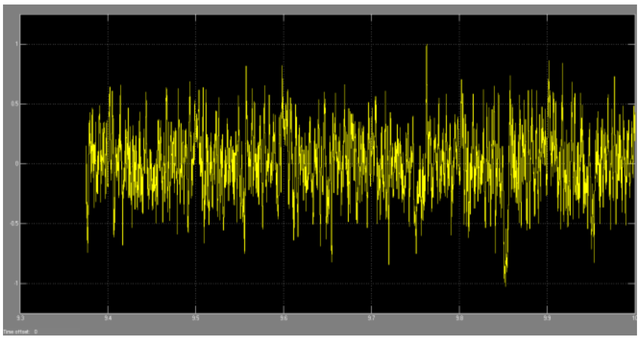

Fig (e) Error Signal

V. CONCLUSION

Adaptive Noise Cancellation is an alternative way of cancelling noise present in a corrupted signal. The principal advantage of the method is in its adaptive capability, its low output noise, and its low signal distortion .It is proved that the proposed NLMS algorithm gives better error performance. The implementation and simulation of Adaptive LMS filter using NLMS algorithm have been done using MATLAB Simulink environment and their response have been studied in waveform in given the simulation result.

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